IN THE CLAIMS

The following claims are presented for examination.

1. (Currently amended) A method for the variable rate coding of a speech signal, comprising the steps of:

(a) classifying the speech signal as either active or inactive;

(b) classifying said active speech into one of a plurality of types of active speech;

(c) selecting an encoder mode from a plurality of parallel encoder modes, wherein selecting the encoder mode is based on whether the speech signal is active or inactive, and if active, based further on said type of active speech, wherein said plurality of parallel encoder modes comprises a code excited linear prediction (CELP) encoder mode, a prototype pitch period (PPP) encoder mode, and a noise excited linear prediction (NELP) encoder mode; and

(d) encoding the speech signal according to said selected encoder mode, forming an encoded speech signal.

2. (Previously presented) The method of claim 1, further comprising the step of decoding said encoded speech signal according to said selected encoder mode, forming a synthesized speech signal.

3. (Canceled)

4. (Previously presented) The method of claim 3, wherein said step of encoding encodes according to said selected encoder mode at a predetermined bit rate associated with said selected encoder mode.

5. (Previously presented) The method of claim 4, wherein said CELP encoder mode is associated with a bit rate of 8500 bits per second, said PPP encoder mode is associated with a bit rate of 3900 bits per second, and said NELP encoder mode is associated with a bit rate of 1550 bits per second.

6. (Previously presented) The method of claim 3, wherein said plurality of parallel encoder modes further comprises a zero rate mode.

7. (Original) The method of claim 1, wherein said plurality of types of active speech include voiced, unvoiced, and transient active speech.

8. (Previously presented) The method of claim 7, wherein said step of selecting the encoder mode comprises the steps of:

(a) selecting a CELP encoder mode if said speech is classified as active transient speech;

(b) selecting a PPP encoder mode if said speech is classified as active voiced speech; and

(c) selecting a NELP encoder mode if said speech is classified as inactive speech or active unvoiced speech.

9. (Previously presented) The method of claim 8, wherein said encoded speech signal comprises codebook parameters and pitch filter parameters if said CELP encoder mode is selected, codebook parameters and rotational parameters if said PPP encoder mode is selected, or codebook parameters if said NELP encoder mode is selected.

10. (Original) The method of claim 1, further comprising the step of calculating initial parameters using a "look ahead."

11. (Original) The method of claim 10, wherein said initial parameters comprise LPC coefficients.

12. (Previously presented) The method of claim 1, wherein said plurality of parallel encoder modes comprises a NELP encoder mode, wherein the speech signal is represented by a residual signal generated by filtering the speech signal with a Linear Predictive Coding (LPC) analysis filter, and wherein said step of encoding comprises the steps of:

(i) estimating the energy of the residual signal, and

(ii) selecting a codevector from a first codebook, wherein said codevector approximates said estimated energy;

and wherein said step of decoding comprises the steps of:

(i) generating a random vector,

(ii) retrieving said codevector from a second codebook,

(iii) scaling said random vector based on said codevector, such that the energy of said scaled random vector approximates said estimated energy, and

(iv) filtering said scaled random vector with a LPC synthesis filter, wherein said filtered scaled random vector forms said synthesized speech signal.

13. (Original) The method of claim 12, wherein the speech signal is divided into frames, wherein each of said frames comprises two or more subframes, wherein said step of estimating the energy comprises the step of estimating the energy of the residual signal for each of said subframes, and wherein said codevector comprises a value approximating said estimated energy for each of said subframes.

14. (Original) The method of claim 12, wherein said first codebook and said second codebook are stochastic codebooks.

15. (Original) The method of claim 12, wherein said first codebook and said second codebook are trained codebooks.

16. (Original) The method of claim 12, wherein said random vector comprises a unit variance random vector.

17. (Currently amended) A variable rate coding system for coding a speech signal, comprising:

classification means for classifying the speech signal as active or inactive, and if active, for classifying the active speech as one of a plurality of types of active speech; and

a plurality of parallel encoding means for encoding the speech signal as an encoded speech signal, wherein said parallel encoding means are dynamically selected to encode the speech signal based on whether the speech signal is active or inactive, and if active, based further on said type of active speech, wherein said plurality of parallel encoder means comprises a code excited linear prediction (CELP) encoding means, a prototype pitch period (PPP) encoding means, and a noise excited linear prediction (NELP) encoding means.

18. (Previously presented) The system of claim 17, further comprising a plurality of parallel decoding means for decoding said encoded speech signal.

19. (Canceled)

20. (Previously presented) The system of claim 18, wherein said plurality of parallel decoding means includes a CELP decoding means, a PPP decoding means, and a NELP decoding means.

21. (Previously presented) The system of claim 19, wherein each of said parallel encoding means encodes at a predetermined bit rate.

22. (Original) The system of claim 21, wherein said CELP encoding means encodes at a rate of 8500 bits per second, said PPP encoding means encodes at a rate of 3900 bits per second, and said NELP encoding means encodes at a rate of 1550 bits per second.

23. (Previously presented) The system of claim 19, wherein said plurality of parallel encoding means further includes a zero rate encoding means, and wherein said plurality of parallel decoding means further includes a zero rate decoding means.

24. (Original) The system of claim 17, wherein said plurality of types of active speech include voiced, unvoiced, and transient active speech.

25. (Previously presented) The system of claim 24, wherein said CELP encoding means is selected if said speech is classified as active transient speech, wherein said PPP encoding means is selected if said speech is classified as active voiced speech, and wherein said NELP encoding means is selected if said speech is classified as inactive speech or active unvoiced speech.

26. (Previously presented) The system of claim 17, wherein said encoded speech signal comprises codebook parameters and pitch filter parameters if said CELP encoding means is selected, codebook parameters and rotational parameters if said PPP encoding means is selected, or codebook parameters if said NELP encoding means is selected.

27. (Previously presented) The system of claim 17, wherein the speech signal is represented by a residual signal generated by filtering the speech signal with a Linear Predictive Coding (LPC) analysis filter, and wherein said plurality of parallel encoding means includes a NELP encoding means comprising:

energy estimator means for calculating an estimate of the energy of the residual signal, and

encoding codebook means for selecting a codevector from a first codebook, wherein said codevector approximates said estimated energy;

and wherein said plurality of decoding means includes a NELP decoding means comprising:

random number generator means for generating a random vector,

decoding codebook means for retrieving said codevector from a second codebook,

multiply means for scaling said random vector based on said codevector, such that the energy of said scaled random vector approximates said estimate, and

means for filtering said scaled random vector with an LPC synthesis filter, wherein said filtered scaled random vector forms said synthesized speech signal.

28. (Previously presented) The system of claim 27, wherein the speech signal is divided into frames, wherein each of said frames comprises two or more subframes, wherein said energy estimator means calculates an estimate of the energy of the residual signal for each of said subframes, and wherein said codevector comprises a value approximating said subframe estimate for each of said subframes.

29. (Previously presented) The system of claim 27, wherein said first codebook and said second codebook are stochastic codebooks.

30. (Previously presented) The system of claim 27, wherein said first codebook and said second codebook are trained codebooks.

31. (Previously presented) The system of claim 27, wherein said random vector comprises a unit variance random vector.